**Listing of the Claims:**

The following is a complete listing of all the claims in the application, with an indication of the status of each:

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1 (Previously Amended). A speech coding apparatus including at least:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal, and

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal, comprising:

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a discrimination section for discriminating a voiced sound mode and an unvoiced sound mode on a basis of a past quantized gain of an adaptive codebook;

a sound source quantization section which has a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination section, and searches combinations of code vectors stored in said codebook and a plurality of shift amounts used to shift positions of the pulses so as to output a combination of a code vector and shift amount which minimizes distortion relative to input speech; and

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section.

2 (Previously Amended). A speech coding apparatus including at least:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal, and

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal, comprising:

a discrimination section for discriminating a voice soundmode and an unvoiced sound mode on a basis of a past quantized gain of an adaptive codebook;

a sound source quantization section which has a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination section, and outputs a code vector that minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule; and

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section.

3 (Previously Amended). A speech coding apparatus including at least:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay an a gain from a pst quanized sound source signal by using an adaptive codebook, and obtaining a reidue by predicting a speech signal, and

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal, comprising:

a discrimination section for discriminating a voice sound mode and an unvoiced sound mode on the basis of a past quantized gain of an adaptive codebook;

a sound source quantization section which has a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based an output from said discrimination section, and a gain codebook for quantizing gains, and searches combinations of code vectors stored in said codebook, a plurality of shift amounts used to shift positions of the pulses, and gain code vectors stored in said gain codebook so as to output a combination of a code vector, shift amount, and gain code vector which minimizes distortion relative to input speech; and

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section.

4 (Previously Amended). A speech coding apparatus including at least:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay an a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal, and

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal, comprising:

a discrimination section for discriminating a voice sound mode and an unvoiced sound mode on the basis of a past quantized gain of an adaptive codebook;

a sound source quantization section which has a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination section indicates a predetermined mode, and a gain codebook for quantizing gains, and outputs a combination of a code vector and gain code vector which minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule; and

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section.

5 (Canceled).

6 (Previously Amended). A speech coding/decoding apparatus comprising:

a speech coding apparatus including:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal,

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal,

a discrimination section for discriminating a voice sound mode and

an unvoiced sound mode on the basis of a past quantized gain of a adaptive codebook, and

a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses when an output from said discrimination section indicates a predetermined mode,

said sound source quantization section searching combinations of code vectors stored in said codebook and a plurality of shift amounts used to shift positions of the pulses so as to output a combination of a code vector and shift amount which minimizes distortion relative to input speech, and further including

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codeboook section, and an output from said sound source quantization section; and

a speech decoding apparatus including at least:

a demultiplexer section for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information,

a mode discrimination section for discriminating a mode by using a past quantized gain in said adaptive codebook,

a sound source signal reconstructing section for reconstructing a sound source signal by generating  non-zero pulses from the quantized sound source information when an output from said discrimination indicates a predetermined mode, and

a synthesis filter section which is constituted by spectrum parameters and reproduces a speech signal by filtering the sound source signal.

7 (Previously Amended). A speech coding/decoding apparatus comprising:

a speech coding apparatus including:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal,

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal,

a discrimination section for discriminating a voice sound mode and an unvoiced sound mode on the basis of a past quantized gain of an adaptive codebook, and

a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination section,

said sound source quantization section outputting a combination of a code vector and shift amount which minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule, and further including

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section; and

a speech decoding apparatus including at least:

a demultiplexer section for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information,

a mode discrimination section for discriminating a mode by using a

past quantized gain in said adaptive codebook,

a sound source signal reconstructing section for reconstructing a sound source signal by generating positions of pulses according to a predetermined rule and generating amplitudes or polarities for the pulses from a code vector when an output from said discrimination section indicates a predetermined mode, and

a synthesis filter section which includes spectrum parameters and reproduces a speech signal by filtering the sound source signal.

8 (Previously Amended). A speech coding apparatus comprising:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter;

means for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal; and

mode discrimination means for receiving a past quantized adaptive codebook gain and performing mode discrimination associated with a voiced/unvoiced mode by comparing the gain with a predetermined threshold, and

further comprising:

sound source quantization means for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the signal, and searching combinations of code vectors stored in a codebook for collectively quantizing amplitudes or polarities of a plurality of pulses in a predetermined mode and a plurality of shift amounts used to temporally shift a predetermined pulse position so as to select a combination of an index of a code vector and a shift amount which minimizes distortion relative to input speech;

gain quantization means for quantizing a gain by using a gain

codebook; and

multiplex means for outputting a combination of outputs from said spectrum parameter calculation means, said adaptive codebook means, said sound source quantization means, and said gain quantization means.

9 (Original). An apparatus according to claim 8, wherein said sound source quantization means uses a position generated according to a predetermined rule as a pulse position when mode discrimination indicates a predetermined mode.

10 (Original). An apparatus according to claim 9, wherein when mode discrimination indicates a predetermined mode, a predetermined number of pulse positions are generated by random number generating means and output to said sound source quantization means.

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11 (Original). An apparatus according to claim 8, wherein when mode discrimination indicates a predetermined mode, said sound source quantization means selects a plurality of combinations from combinations of all code vectors in said codebook and shift amounts for pulse positions in an order in which a predetermined distortion amount is minimized, and outputs the combinations to said gain quantization means, and

said gain quantization means quantized a plurality of sets of outputs from said sound source quantization means by using said gain codebook, and selects a combination of a shift amount, sound source code vector, and gain code vector which minimizes the predetermined distortion amount.